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Patentanmeldung Nr. / Patent application No. / Demande de brevet n°

00200055.2

Der Präsident des Europäischen Patentamts;
Im Auftrag

For the President of the European Patent Office

Le Président de l'Office européen des brevets
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I.L.C. HATTEN-HECKMAN

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Blatt 2 der Bescheinigung
Sheet 2 of the certificate
Page 2 de l'attestation

Anmeldung Nr.:
Application no.:
Demande n°:
00200055.2

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Anmelder:
Applicant(s):
Demandeur(s):
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Generation of coefficients for prediction filter in encoder

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Remarques:

Generation of Coefficients for prediction filter in encoder.

The invention relates to a transmitter for transmitting a digital information signal via a transmission medium, including:

- input means for receiving the digital information signal,
- adaptive prediction filter means adapted to derive a prediction signal from the digital information signal in dependence of an array of prediction filter coefficients,
- first signal combination means for combining the digital information signal and said prediction signal so as to obtain a residual signal,
- encoding means encoding for encoding said residual signal so as to obtain an encoded signal,
- coefficient generator means for generating an array of filter coefficients A[i] in response to the digital information signal, i being an integer for which holds $0 \leq i \leq p$ and whereby p is a variable,
- output means for supplying the encoded signal to an output terminal for transmission via a transmission medium.

The invention further relates to a receiver for receiving a transmission signal and generating a digital information therefrom, and to a transmission method.

A transmitting device and receiving device of the type defined in the opening paragraph is known from J. Audio Eng. Soc., Vol. 44, No. 9, pp. 706 – 719, September 1996. The known transmitting device is intended for efficiently reducing the bit rate for the transmission of a digital information signal. Therefore, prior to encoding the digital information signal a predicted version of the digital information signal is subtracted from the digital information signal. The thus obtained residual signal is successively encoded in the encoder and transmitted via the transmission medium. The performance of the linear prediction filter is crucial for the coding gain of the encoder. The performance of the linear prediction filter is determined by the prediction filter coefficient. A common method of finding the prediction coefficients (a-parameters) is the auto correlation method. It appeared that a-parameters determined with the auto correlation method did not result in the optimum coding gain.

The invention aims at providing a transmitting device and receiving device having a more efficient method of transmitting and receiving the digital information.

The transmitter in accordance with the invention is characterized in that the device further comprises

- smoothing means for smoothing the array of coefficients A[i] so as to obtain the array of prediction coefficients for supplying to the adaptive prediction filter means.

The receiver in accordance with the invention is characterized in that the receiver further comprises;

- smoothing means for smoothing the array of coefficients A[i] so as to obtain the array of prediction coefficients for supplying to the adaptive prediction filter means.

The invention is based on the following recognition. As long as transmission bandwidth and storage capacity is limited, there will be need to increase the bit reduction of digital signals. It appeared that the acquired prediction filter coefficients (a-parameters) obtained by for example the auto-correlation method, does not result in an optimal coding gain. During extensive searches on the a-parameter values, it appeared that better coding gains could be achieved by slightly modified a-parameters. Coding gain improvements of around 4% could be achieved. However, the complexity needed in the transmitting device for such an extensive search would be enormous, and thus not practical.

After investigation of the difference between the a-parameters before and after the extensive search, it appeared that the a-parameters are roughly the same (see Figure 8). When zooming in at the first few a-parameters the a-parameters obtained by the auto-correlation method show a rippling behavior while the a-parameters resulting from the extensive search are a smoothed version of it (see Figure 9).

Instead of executing an extensive search on the a-parameters so as to obtain the prediction filter coefficients, the a-parameters acquired by the auto correlation method are post-processed by a smoothing function. The improvement of the coding gain coding gain obtained with the smoothing function is comparable with the coding gain obtained with extensive search. Figure 10 shows the results for a 5-minute DSD fragment, where the result was even an improvement of 4.5%. Figure 11 shows that the performance is comparable of both methods (the extensive search would gain another 0.4% for this fragment). The increase of the complexity of the arrangement to implement the smoothing method is negligible

compared with the increase of the complexity of the arrangement when implementing the extensive search method.

These and other objects of the invention will become apparent from and elucidated further with reference to the embodiments described in the following figure description in which

figure 1 shows an embodiment of an SACD coder,

figure 2 shows the coding gain of successive frames,

figure 3:shows the coding gain of successive frames and the corresponding value of max $|a_i|$,

figure 4 shows the coding gain as a function of max $|a_i|$,

figure 4a shows a detailed view of the coding gain as a function of max $|a_i|$ shown in figure 4,

figure 5 and 5a shows the coding gain and impulse response as a function of the prediction order,

figure 6 and 6 a shows the coding gain and impulse response as a function of the prediction order after applying the first two steps of the method in accordance to the invention,

figure 7 shows the coding gain of the same successive frames in figure 2 after applying the method in accordance to the invention,

figure 7a shows the value of max $|a_i|$ of successive frames after applying the method in accordance to the invention,

figure 7b shows the coding gain as a function of the max $|a_i|$ after applying the method in accordance to the invention,

figure 8 shows an the arrays of prediction filter coefficient before and after extensive search,

figure 9 shows a detail of the arrays of prediction filter coefficients before and after the extensive search,

figure 10 shows the coding gain obtained by using the first array of coefficients versus the coding gain obtained by using the smoothed version of the first array of coefficients,

figure 11 shows the coding gain obtained by using the array of coefficients obtained by extensive search versus the coding gain obtained by using the smoothed version of the first array of coefficients,

figure 12 shows an embodiment of a transmitter in accordance with the invention,.and

figure 13 shows an embodiment of a receiver in accordance with the invention.

Introduction

For "Super Audio CD" (SACD), the audio signals are lossless coded using framing, linear prediction and entropy coding. The coding gain is heavily determined by the quality of prediction. It is therefore important to have an algorithm that optimally determines the prediction coefficients. Figure 1 shows a block diagram of a simplified scheme that is used in a SACD encoder.

Disclosed is a low-complexity method for generating prediction coefficients that result in an improved coding gain.

The problem is that frames, for which high coding gain can be expected, often suffer from significantly low gains. Such a low coding gain on frames brings the average coding gain lower, consequently having adverse implication on the storage capacity. Figure 2 shows the coding gain of successive frames illustrating the problem. In an excerpt when the coding gain is relatively high (average 2.7), we see clearly that there are frames with a very low coding gain (close to 1.0).

Interactively, selecting a lower prediction order would have avoided this problem. However, an important issue is that coding gain is only available once the entire coding process has taken place and the complexity of interactively/iteratively re-encoding the frame using lower order is high.

Disclosed Solution

Here, we disclose a detection mechanism, which is capable of signaling LPC coefficients rendering low coding gain, where higher gain was possible, prior to encoding of the entire frame. The detection is based on the dynamic range of the direct form prediction-filter coefficients, often referred to as a-parameters. Consider Figure 3, where the top trace is same as figure 2 and the lower trace shows the value of the maximum of the absolute values of the a-parameters. It is clear from this Figure that the occurrence of low coding gain has a correlation with the occurrence of high values of max $|a_i|$. Based on this detection, a 3-stage process is disclosed to eliminate this problem. The result is prediction coefficients offering higher performance of the encoding process.

In Figure 4, we illustrate a slightly different version of Figure 2. Here, the maximum of the absolute values of a-parameters, max $|a_i|$, is shown on the horizontal axis and the vertical axis shows the corresponding coding gain. The a-parameters are determined by using the auto-correlation method in combination with the Schur algorithm. A line is drawn at a value of 10. Notice that for cases, where the maximum of the absolute value of a-parameters is higher than 80, there isn't a single occurrence of a frame with high coding gain.

If we zoom in to one of the "problem frames", i.e. a frame with low coding gain and look at the prediction gain as a function of the prediction order, then we see Figure 5. Figure 5 and 5a highlights two issues. Figure 5 shows the coding gain as a function of the prediction order. Figure 5a shows the impulse response for various prediction orders. One may conclude from figure 5 that a higher coding gain was possible, for instance at prediction order 80, the gain was maximum. The second issue is highlighted in Figure 5a. The impulse response only changes in scale but not in shape. It is interesting to interpret this in the spectral domain, if one realizes that the prediction gain is proportional to the non-flatness in the spectrum [Flanagan paper]. Thus all the flattening has already been achieved in lower order, when the order is increased, only the scale is changing while the spectral coloring (or the flattening in the inverse filtered domain) is not changing. This points to some ill conditioning. Next, we discuss the steps that we have taken to avoid this problem by inspecting the impulse response of the prediction filter, i.e. without having to go through the entire coding process. We observed that that there is a relation between low coding gains and the maximum absolute value of the a-parameters returned by the auto-correlation method. For frames with a good coding gain the value of the maximum absolute a-parameters is always below 10, and problem frames show values of 100 or even 600.

Step 1: Detection of Problem Frames

The first step in the method in accordance to the invention is the detection of the problem frames, i.e., frames for which the coding gain is low, while a higher gain was possible. Accordingly, Step 1 includes the computation of the a-parameters based on the auto-correlation method. Next, a value T, representing the absolute maximum value of a_i is computed on the basis of the a-parameters, $A = \{ a_1, a_2, ..., a_{128} \}$, such that, $T = \max |A|$. A frame is declared as a problem frame if T is greater than a predefined value, for example 10.0.

Step 2: Noise Addition

Some of these problems are eliminated if one adds a very low-level random noise signal to the input prior to LPC analysis. Equivalently, the value of $R[0]$, the first auto-correlation coefficient, can be raised by a very small amount. If the threshold of step 1 is exceeded, the auto-correlation function coefficient $R[0]$ is modified such that

$$R'[0] = R[0] * (1.0 + 4.10^{-6}),$$

and the a-parameters are recalculated in accordance with the new value of $R[0]$.

Although such a minor modification solves the problem in most cases, there were frames for which the problem persisted, or even got worse (in terms of coding gain).

Step 3: Prediction-Order Reduction

The maximum absolute value of the now acquired a-parameters is again a good detection mechanism to see if problems are to be expected. For these frames where the addition of a little noise doesn't solve the problem, another measure must be taken and that is to take a reduced order for LPC prediction. It is difficult, however, to select the optimal prediction order. A threshold that seems to achieve good results is:

$$\text{New Order} = \min\left(\left(\textit{current order} * 8 / 9\right), 80 \right)$$

where the *current order* is the order of prediction with which the analysis has been done.

In practice it appeared that with the combination of these three steps (detection via max(abs(a-parameters)), noise addition, and order reduction), the coding gain improved considerably for problem frames. More importantly, the good frames were not effected adversely with this mechanism.

Results

Figure 6, shows the improvements that have been achieved after applying the first 2 steps of the procedure discussed earlier. This is in contrast with Figure 5 where the coding gain dropped sharply beyond 80th-order prediction. Now the a-parameters also exhibit a reasonable behavior in terms of the amplitude dynamics. As indicated earlier, even after applying the 2 steps, some problem frames persist and the application of step 3 eliminates them as well.

Finally, Figure 7, 7a and 7b, shows clearly that the bad frames have been completely eliminated.

We have discussed a three-stage approach to eliminate the bad frames entirely. This has been achieved by conditioning the signal in such a way that they deliver better prediction coefficients.

While, only Schur recursion is discussed in this paper, experiments show that identical problems exist when using the Cholesky decomposition. Also here, the disclosed solution helps to completely eliminate the ill-conditioned a-parameters.

"Smoothing" of the a-parameters could deliver (about) the same results as the extensive search, but for a negligible complexity.

A smoothing algorithm uses the a-parameters to calculate a new set of a-parameters. Possible implementations are could be done by for example FIR or IIR-filtering. After some experiments a variant of the IIR-method was used selected:

```
/* Apar[0] .. Apar[po-1] contain the a-parameters of the filter with prediction order po */
for (i = 1; i < po – 1; i++)
{
        Apar[i] = Apar[i-1] + 2*Apar[i] + Apar[i+1]
}
```

As can be seen from this part of code, the algorithm is recursive because the value that is adapted is reused in the next calculation. It appeared that it worked better than the non-recursive approach.

On average some 3% coding gain improvement can be reached when applying this smoothing to the a-parameters. Figure 10 shows the results for a 5-minute DSD fragment, where the result was even an improvement of 4.5%. The complexity of this method is negligible. As can be seen from Figure 10 there were no frames where smoothing decreased the coding gain. Comparable results were obtained for some other DSD-fragments, with the exception that for a very few high gain frames (>3) the gain showed a minor decrease.

A comparison is made between the coding gains resulting from the extensive search and the ones resulting from smoothing of the a-parameters that came out of the auto correlation method. Figure 11 shows that the performance is comparable of both methods (the extensive search would gain another 0.4% for this fragment. To decrease the complexity of the arrangement it is obvious that the method using smoothing of the a-parameters is preferred.

The disclosed method is also lower in complexity when compared to the alternative that the entire coding process has to be performed before the coding gain is known.

Whilst the invention is described with reference to preferred embodiments thereof, it is to be understood that these are not limitative examples. Thus various modifications may become apparent to those skilled in the art, without departing from the scope of the invention, as defined by the claims.

The word 'comprising' does not exclude the presence of other elements or steps than those listed in a claim. Any reference signs do not limit the scope of the claims. The invention can be implemented by means of both hardware and software. Several "means may be represented by the same item of hardware. Further the invention lies in each and every novel feature or combination of features.

CLAIMS:

1. A transmitting device for transmitting a digital information signal via a transmission medium, including:
- input means for receiving the digital information signal,
- adaptive prediction filter means adapted to derive a prediction signal from the digital information signal in dependence of an array of prediction filter coefficients,
- first signal combination means for combining the digital information signal and said prediction signal so as to obtain a residual signal,
- encoding means encoding for encoding said residual signal so as to obtain an encoded signal,
- coefficient generator means for generating an array of filter coefficients A[i] in response to the digital information signal, i being an integer for which holds $0 \leq i \leq p$ and whereby p is a variable,
- output means for supplying the encoded signal to an output terminal for transmission via a transmission medium,

characterized in that the device further comprises
- smoothing means for smoothing the array of coefficients A[i] so as to obtain the array of prediction coefficients for supplying to the adaptive prediction filter means.

2. Transmitter as claimed in claim 1, characterized in that the smoothing means comprises low pass filtering means for low pass filtering of the coefficients so as to obtain the coefficient signal.

3. Transmitter as claimed in claim 2, characterized in that the low pass filtering means are in the form of a FIR filter.

4. Transmitter as claimed in claim 2, characterized in that the low pass filtering means are in the form of an IIR filter.

5. Transmitter as claimed in any of the preceding claim in the form of an arrangement for writing the encoded signal on a record carrier.

6. Method of transmitting a digital information signal via a transmission medium, comprising the following steps:
- receiving the digital information signal,
- deriving a prediction signal from the digital information signal in dependence of an array of prediction filter coefficients,
- combining the digital information signal and said prediction signal so as to obtain a residual signal,
- encoding said residual signal so as to obtain an encoded signal,
- generating an array of coefficients A[i] in response to the digital information signal, i being an integer for which holds $0 \leq i \leq p$, whereby p is a variable,
- supplying the encoded signal to an output terminal for transmission via a transmission medium,
characterized in that the method further comprises the step
- smoothing the array of coefficients A[i] so as to obtain the array of prediction filter coefficients.

7. Receiver for receiving a transmission signal and generating a digital information therefrom, the receive comprising:
- receiving means for receiving the transmission signal and retrieving an encoded signal therefrom,
- decoding means for decoding the encoded signal so as to obtain a residual signal,
- adaptive prediction filter means adapted to derive a prediction signal from the digital information signal in dependence of an array of prediction filter coefficients,
- signal combination means for combining the residual signal and the prediction signal so as to obtain the digital information signal,
- coefficient generator means for generating an array of filter coefficients A[i] in response to the digital information signal, i being an integer for which holds $0 \leq i \leq p$ and whereby p is a variable,
characterized in that the receiving device further comprises
- smoothing means for smoothing the array of coefficients A[i] so as to obtain the array of prediction coefficients for supplying to the adaptive prediction filter means.

ABSTRACT:

A transmitter is disclosed for transmitting a transmission signal via a transmission medium. The transmitter derives a prediction signal from the digital information signal in dependence of an array of prediction filter coefficients. The array of prediction filter coefficients has been obtained by smoothing a first array of coefficients. Said first array of coefficients has been generated in response to the digital information signal. A residual signal has been obtained by combination of the digital information signal and the prediction signal. The residual signal is encoded so as to obtain an encoded signal. The encoded signal is transmitted via the transmission medium.
(Fig. 1)

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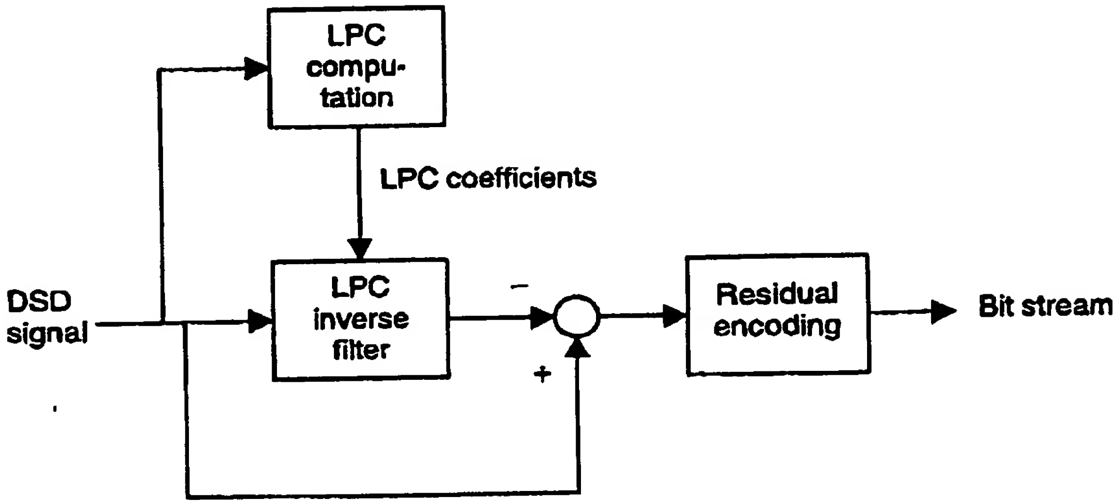
LPC computation
LPC coefficients
DSD signal
LPC inverse filter
−
+
Residual encoding
Bit stream

FIG. 1

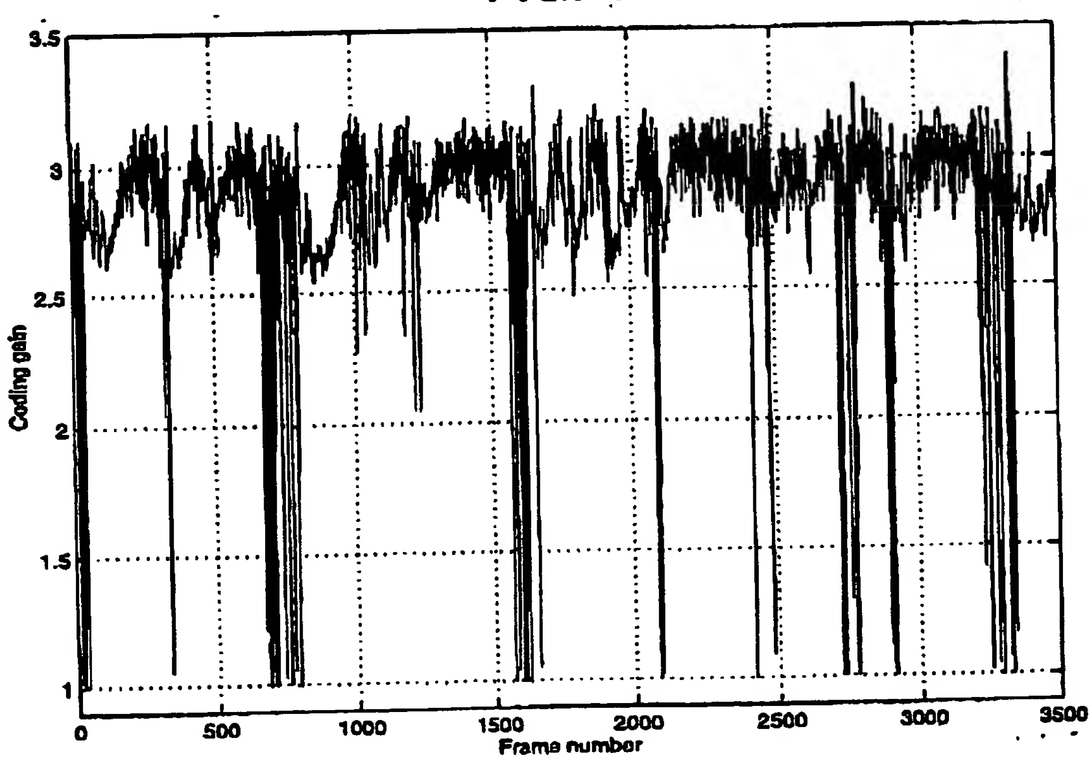
Coding gain
Frame number
3.5
3
2.5
2
1.5
1
0
500
1000
1500
2000
2500
3000
3500

FIG. 2

2/7

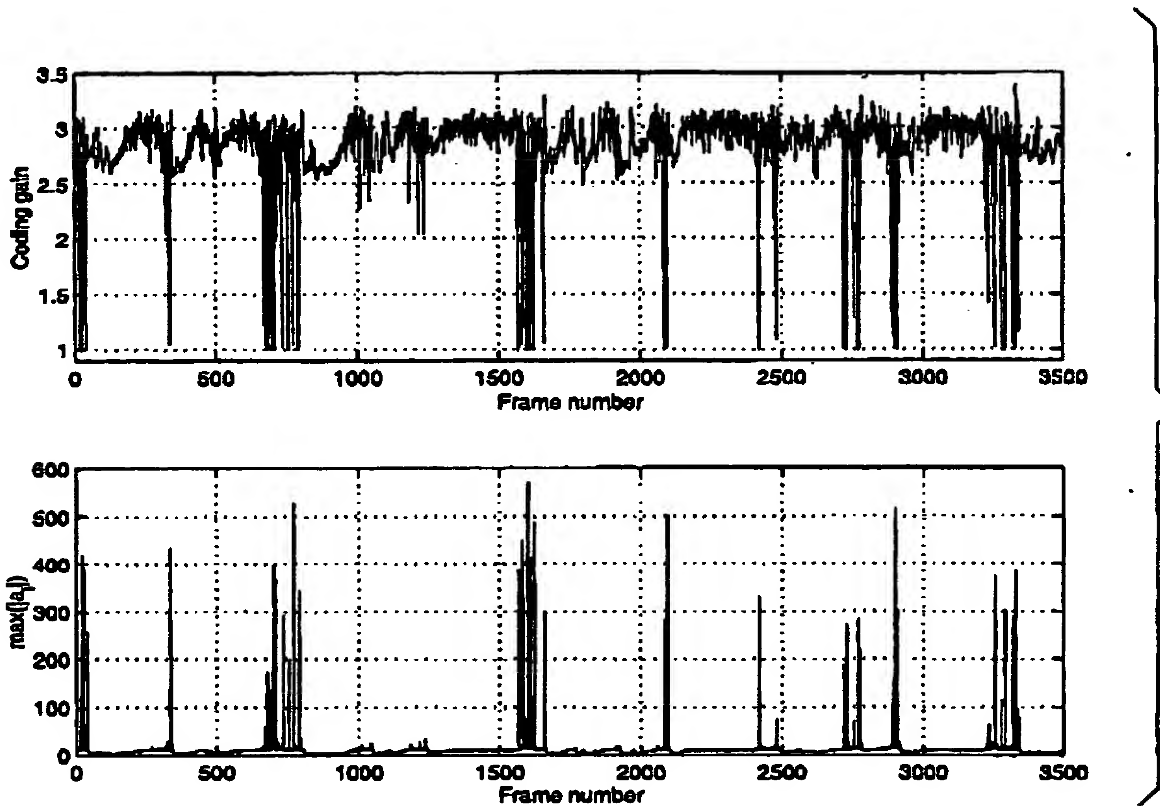

FIG. 3

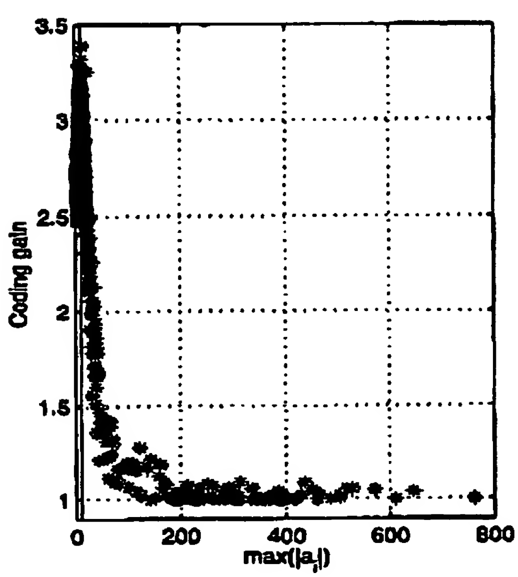

FIG. 4

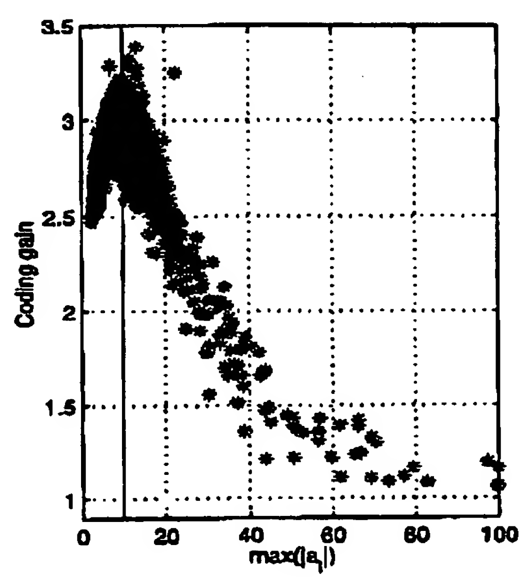

FIG. 4a

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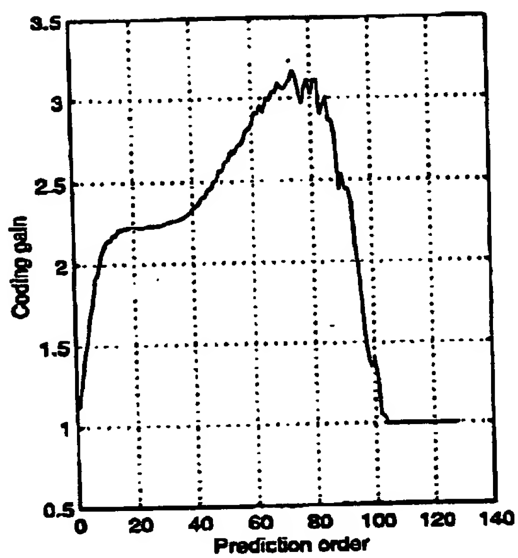

FIG. 5

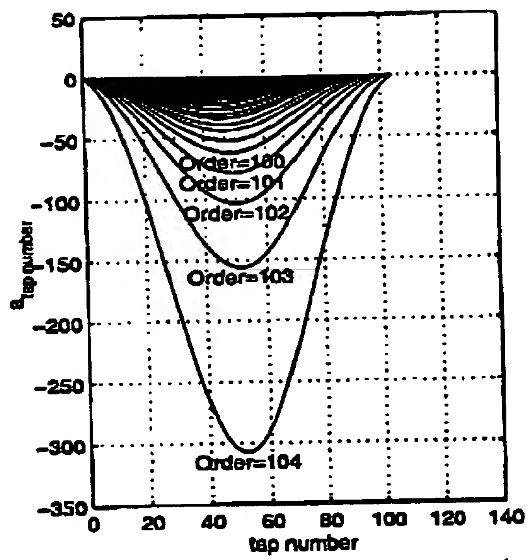

FIG. 5a

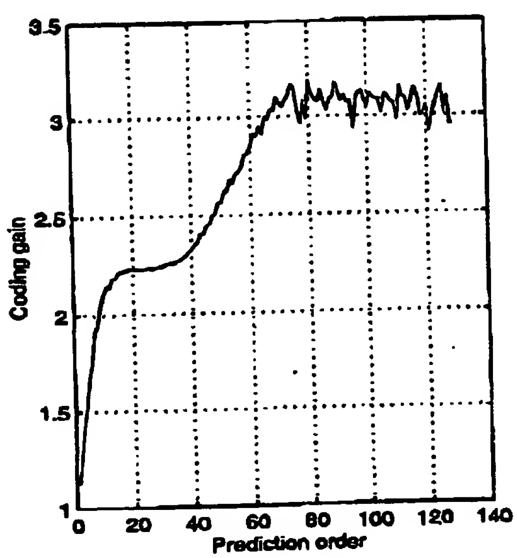

FIG. 6

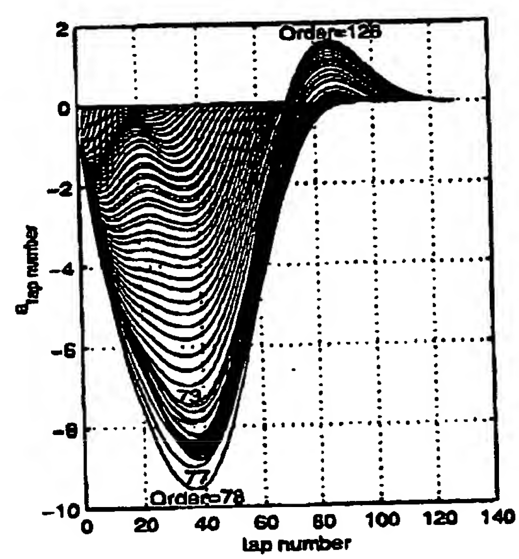

FIG. 6a

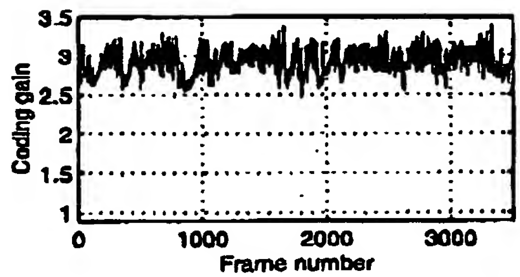

FIG. 7

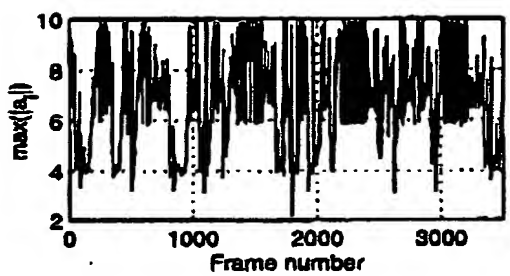

FIG. 7a

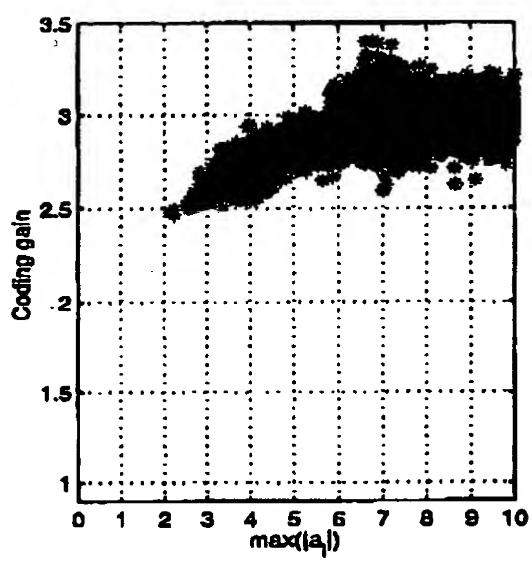

FIG. 7b

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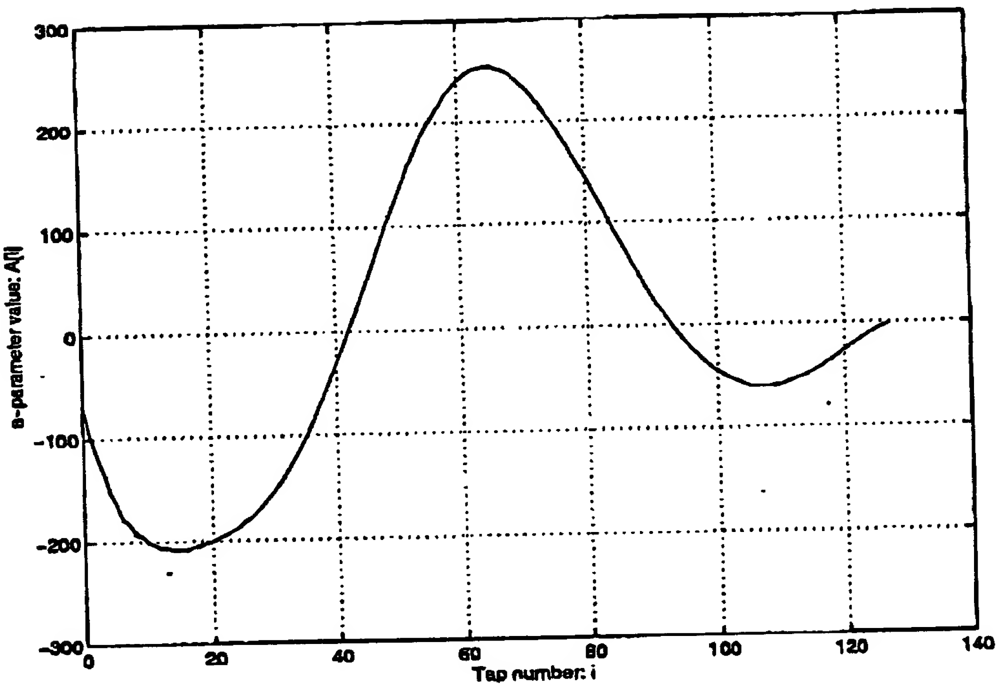

FIG. 8

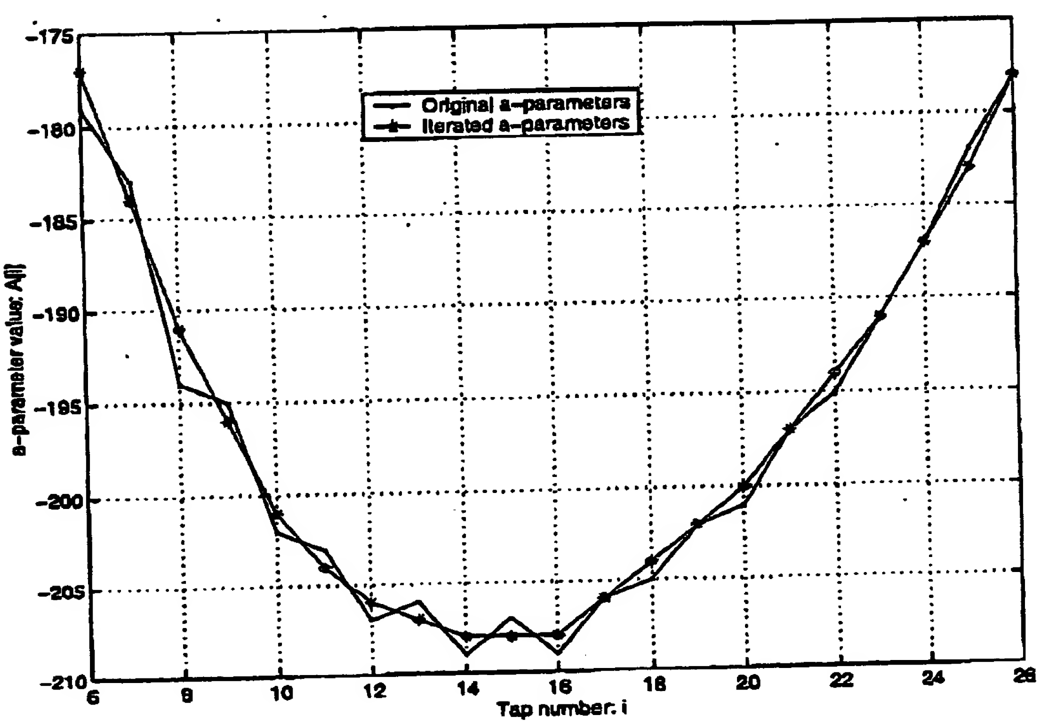

FIG. 9

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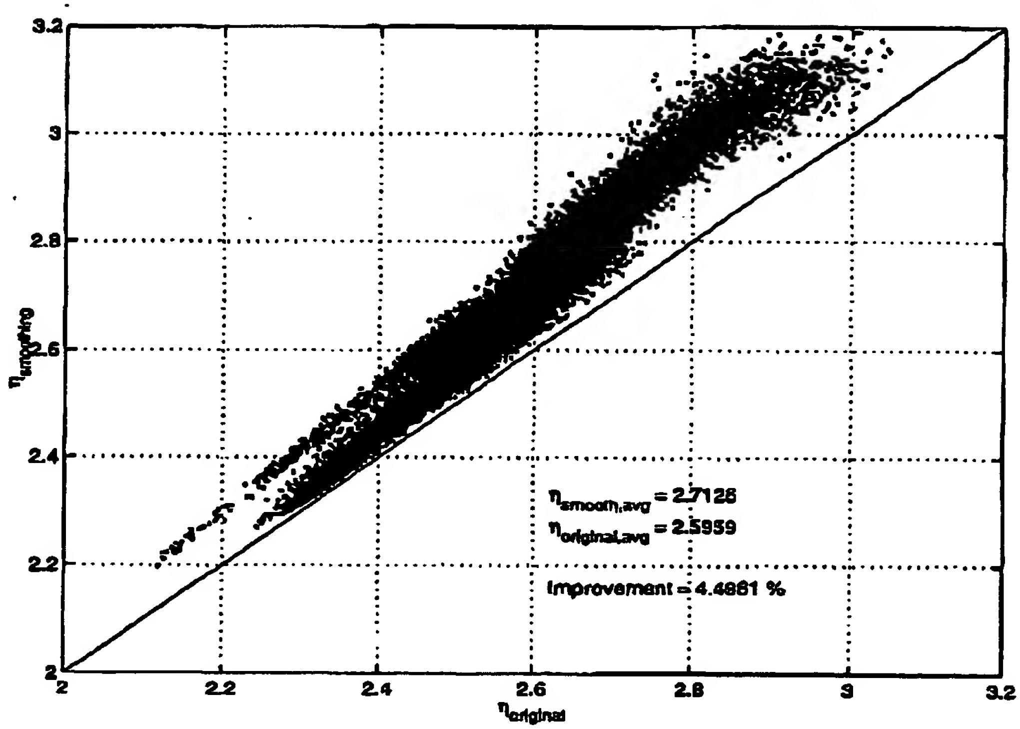

FIG. 10

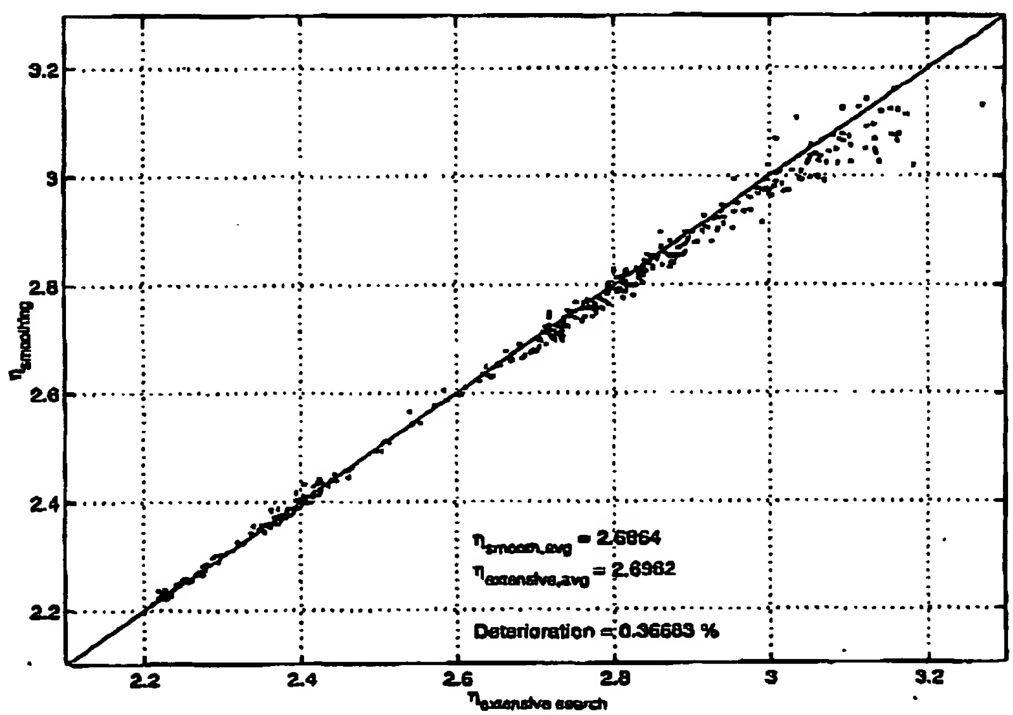

FIG. 11

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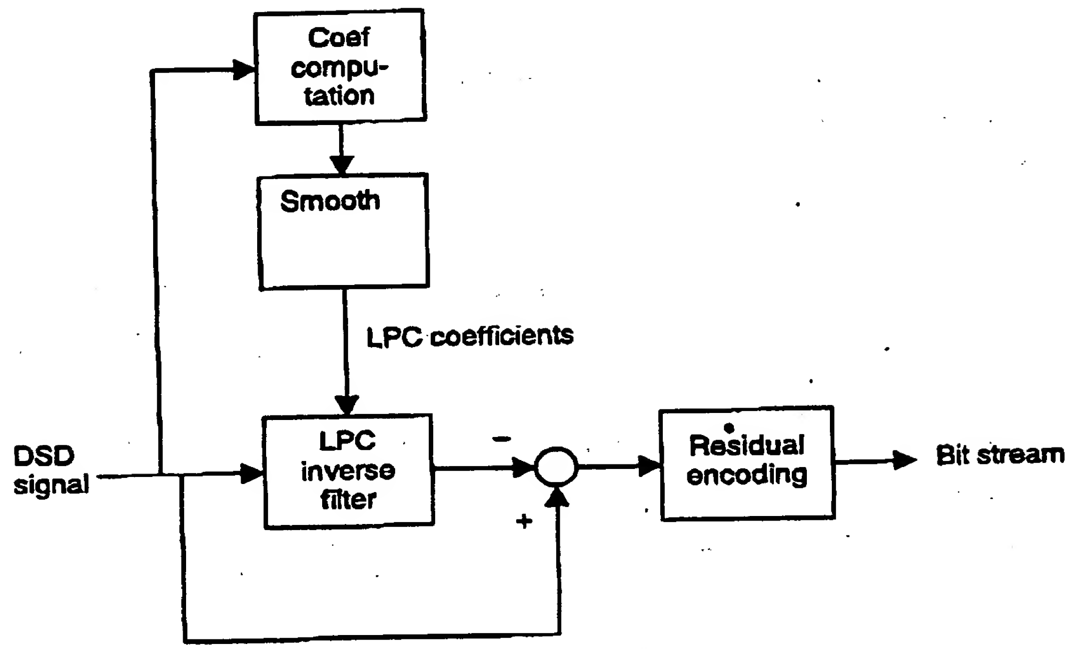
Coef
compu-
tation
Smooth
LPC coefficients
DSD
signal
LPC
inverse
filter
–
+
Residual
encoding
Bit stream

FIG. 12

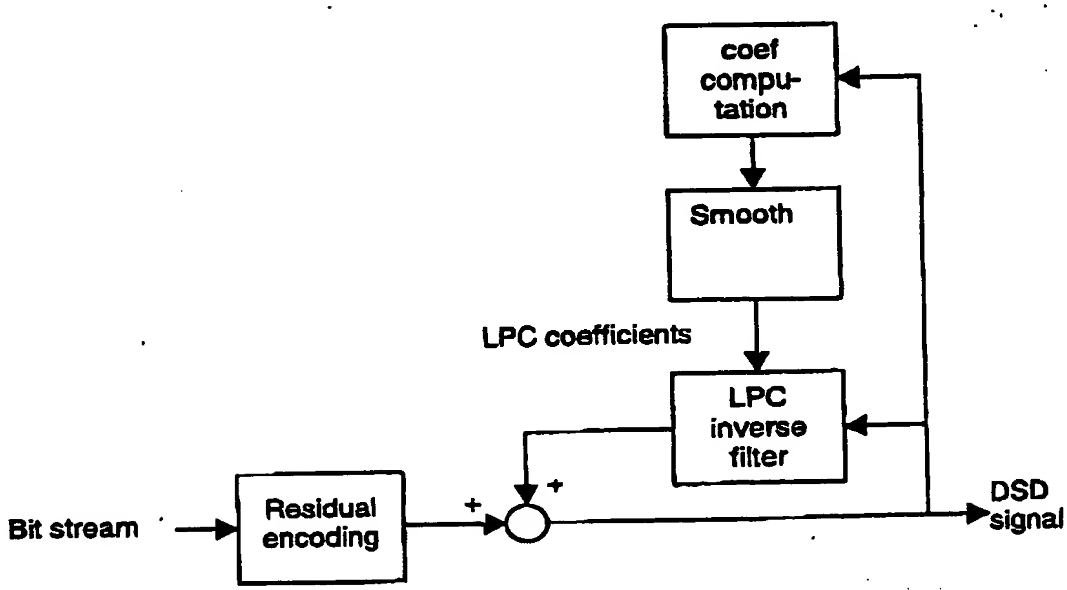
coef
compu-
tation
Smooth
LPC coefficients
LPC
inverse
filter
+
+
Bit stream
Residual
encoding
DSD
signal

FIG. 13